International Journal of Research and Applications
April - June © 2014 Transactions; 1(2): 58-62
ISSN (online): 2349-0020
http://ijraonline.com
ELECRONICS
ORIGINAL ARTICLE

Channel Estimation using Modified Extended Kalman Filter Based Algorithm for Fading Channels

G.Rajender[1], T.AnilKumar[2] and K.Srinivasa Rao[3]
[1 3] Department of ECE, TRR College of Engineering, Hyderabad, Telangana, India -502 319
[2]Department of ECE, SR Engineering College, Warangal, Telangana, India -506 371
[1]rajender427@gmail.com, tvakumar2000@yahoo.co.in and [3]principaltrr@gmail.com

ABSTRACT

The process of characterizing the effect of the physical channel on the input sequence is known as the Channel estimation. The Channel estimation techniques offer low complexity and better performance and are effectively used in communication systems. But they are also wasteful of bandwidth since they use training sequences to estimate the channel, so here; a method is used where the limited length of training sequence is transmitted. The Kalman based algorithm which can be used efficiently for channel estimation procedure. The kalman based algorithm channel estimator leads to a significant gain in performance as compared to the data-only estimator. This algorithm also allows us to predict the state of the system before the frame is actually received. In this paper, the channel estimation is done using Kalman based algorithm to predict the estimates of the state of system. Also the total harmonic distortion of the updated state is calculated and limited within a particular value. The channel under consideration is a Rayleigh fading channel.

Keywords- Channel estimation, Kalman Filter, Rayleigh fading, Total Harmonic Distortion

I. INTRODUCTION

Kalman based algorithm is used in this paper for channel estimation. This process needs equations for the Kalman based algorithm, which are explained in [1]. Channel estimation is an important consideration when we transmit signals through wireless paths. If the channel is assumed to be linear, the channel estimate is simply the estimate of the impulse response of the system. It must be stressed once more that channel estimation is only a mathematical representation of what is truly happening. A good channel estimate is one where some sort of error minimization criteria is satisfied (e.g. MMSE). The process of channel estimation is described in detail in [2]. The input given is a sine wave.

The methods by which different estimation techniques are applied to the channel are shown in the signal processing fields in [3] and [4]. The algorithm is an extension of the Kalman filter. So to know about EKF, it is inevitable to understand Kalman filter, which is in [5]. The MATLAB implementations of Kalman filter are in [6]. A comparison between Kalman based algorithm and Kalman filter is necessary to understand the similarity and differences to find the exact one to be implemented for our purpose. This is explained in [7], where we have the equations for the filters. The predicted estimates and updated estimates are found using kalman based algorithm estimation. This method of estimation is mainly used for the channel estimation and analysis of non-linear-systems. Total Harmonic Distortion between the updated estimate and the actual state is also calculated. The total harmonic distortion or THD of a signal is a measurement of the harmonic distortion present in the signal.

The channel under consideration is a Rayleigh fading channel. Rayleigh fading is a reasonable model when there are many objects in the environment that scatter the radio signal before it arrives at the receiver [8].

II. SYSTEM MODEL

A. *Channel Estimation*

The process of channel estimation has been explained. There are two methods for channel estimation. They are training sequence / pilot based channel estimation and blind channel estimation [2]. Estimation algorithms aim at minimizing the mean squared error.

We try to estimate h in the presence of noise and model mismatch, by only observing the channel output y(n):$y(n)=h^T*x(n)+v(n)$

In the Fig. 1, e(n) is the estimation error. The aim of most channel estimation algorithms is to minimize the mean squared error (MMSE), $E[e^2(n)]$ while utilizing as little computational resources as possible in the estimation process.

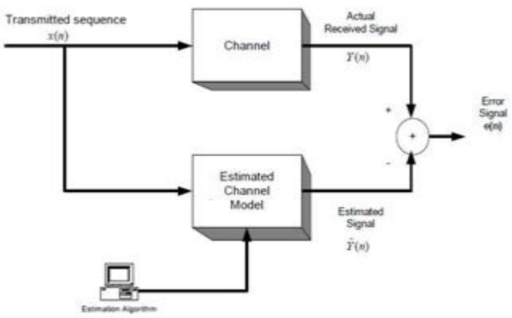

Fig 1. General Channel Estimation

Once an estimation model has been established, its parameters need to be continuously updated (estimated) in order to minimize the error as the channel changes. If the receiver has a priori knowledge of the information being sent over the channel, it can utilize this knowledge to obtain an accurate estimate of the impulse response of the channel. This method is simply called training sequence based Channel estimation. It has the advantage of being used in any radio communications system quite easily.

Even though this is the most popular method in use today, it still has its drawbacks. One of the obvious drawbacks is that it is wasteful of bandwidth. Precious bits in a frame that might have been otherwise used to transport information are stuffed with training sequences for channel estimation. This method also suffers due to the fact that most communication systems send information lumped frames. It is only after the receipt of the whole frame that the channel estimate can be extracted from the embedded training sequence. For fast fading channels this might not be adequate since the coherence time of the channel might be shorter than the frame time.

Two parameters are important as a measure of the quality of estimation. They are bias and variance of estimation. A bias (unknown) means a permanent (un-removable) error in the estimates. An unbiased estimator is desirable. Variance of estimations gives the possible amount of deviation of the estimator from the actual value to be estimated. An estimator with a low variance (ideally minimum) is desirable. Moreover, a linear estimator is desirable due to the ease of implementation. Hence, a linear unbiased estimator with minimum variance is highly desirable.

The pilot based channel estimation is of two types. They are the block type and the comb type. In the block type, all subcarriers in the frame are pilot tones. Also, the frames are transmitted periodically. It uses the same channel estimation for whole In the case of comb type, only some of the subcarriers in the frame are pilot tones. Blind methods on the other hand require no training sequences. They utilize certain underlying mathematical information about the kind of data being transmitted. These methods might be bandwidth efficient but still have their own drawbacks. They are notoriously slow to converge. Their other drawback is that these methods are extremely computationally intensive and hence are impractical to implement in real-time systems. They also do not have the portability of training sequence-based methods. One algorithm that works for a particular system may not work with another due to the fact they send different types of information over the channel.

B. *Kalman Based algorithm*

The Kalman filter, also known as linear quadratic estimation (LQE), is an algorithm that uses a series of measurements observed over time, containing noise (random variations) and other inaccuracies, and produces estimates of unknown variables that tend to be more precise than those based on a single measurement alone. More formally, the Kalman filter operates recursively on streams of noisy input data to produce a statistically optimal estimate of the underlying system state. The algorithm works in a two-step process. In the prediction step, the Kalman filter produces estimates of the current state variables, along with their uncertainties. Once the outcome of the next measurement (necessarily corrupted with some amount of error, including

random noise) is observed, these estimates are updated using a weighted average, with more weight being given to estimates with higher certainty.

Most real systems are non-linear. The Kalman Filter for nonlinear models is denoted the Kalman based Filter because it is an extended use of the original Kalman Filter. Kalman based Filter (KF) uses non-linear models of both the process and observation models while the Kalman Filter (KF) uses linear models [1].

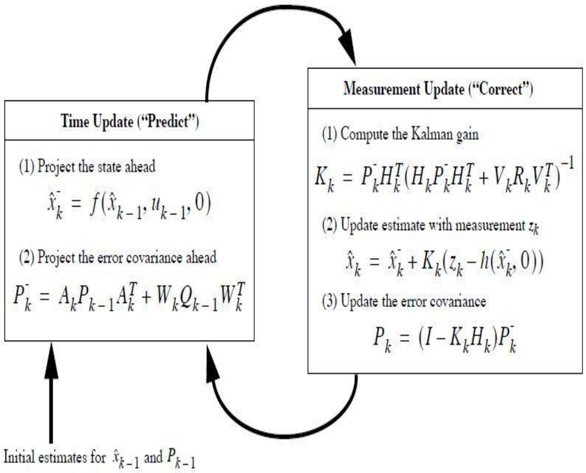

Fig 2. Kalman Filter Prediction Estimation Cycle

Fig 2 shows the Kalman based Filter Prediction Estimation Cycle which gives equations to do the estimation procedure. The Kalman based filter (KF) gives an approximation of the optimal estimate. The non-linearities of the system's dynamics are approximated by a linearized version of the non-linear system model around the last state estimate. For this approximation to be valid, this linearization should be a good approximation of the non-linear model in the entire uncertainty domain associated with the state estimate.

C. *Rayleigh Fading*

Rayleigh fading is the most important type of fading that occurs in the urban environments. It is a statistical model for the effect of a propagation environment on a radio signal, such as that used by wireless devices. Rayleigh fading models assume that the magnitude of a signal that has passed through such a transmission medium (also called a communication channel) will vary randomly, or fade, according to a Rayleigh distribution [8].

The central limit theorem holds that, if there is sufficiently much scatter, the channel impulse response will be well-modelled as a Gaussian process irrespective of the distribution of the individual components. If there is no dominant component the scatter,then such a process will have zero mean phase evenly distributed and between 0 and 2 π radians.

The envelope of the channel response will therefore be Rayleigh distributed. Rayleigh fading is a small-scale effect. There will be bulk properties of the environment such as path loss and shadowing upon which the fading is superimposed. However there will be very many objects around the direct path. These objects may serve to reflect, refract, etc the signal. As a result of this, there are many other paths by which the signal may reach the receiver. When the signals reach the receiver, the overall signal is a combination of all the signals that have reached the receiver via the multitude of different paths that are available. These signals will all sum together, the phase of the signal being important. Dependent upon the way in which these signals sum together, the signal will vary in strength. If they were all in phase with each other, they would all add together. However this is not normally the case, as some will be in phase and others out of phase, depending upon the various path lengths, and therefore some will tend to add to the overall signal, whereas others will subtract. This forms the Rayleigh distribution for the channel.

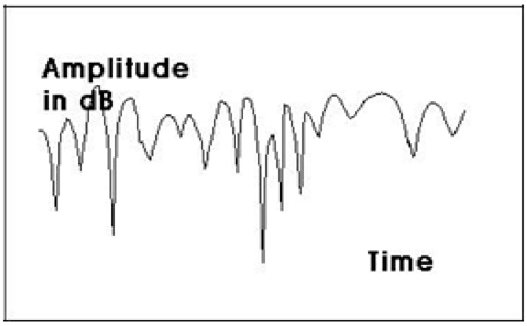

Fig 3. A sample of a Rayleigh fading signal.

Calling the random variable R, it will have a probability density function is as given in the equation below

$$p_R(r) = \frac{2r}{\Omega} e^{-r^2/\Omega}, \; r \geq 0$$

where

$$\Omega = E(R^2)$$

D. *Total Harmonic Distortion*

The total harmonic distortion is defined as the ratio of the sum of the powers of all harmonic components to the power of the fundamental frequency. THD is used to characterize the linearity of audio systems

and the power quality of electric power systems. When the input is a pure sine wave, the measurement is most commonly the ratio of the sum of the powers of all higher harmonic frequencies to the power at the first harmonic, or fundamental frequency.

THD (%) = sqrt (harmonic power) / sqrt (signal power)

THD=harmonic power / signal power

THD is also commonly defined as an amplitude ratio rather than a power ratio, resulting in a definition of THD which is the square root of that given above. The total harmonic distortion, or THD, of a signal is a measurement of the harmonic distortion present and is defined as the ratio of the sum of the powers of all harmonic components to the power of the fundamental frequency.

III. METHODOLOGY

In this project, the Extended Kalman Filter is used for channel estimation. The input given is a sine wave. The channel used is a Rayleigh channel. Then the predicted estimates and updated estimates are to be found. The total harmonic distortion between the updated estimate and the actual state also needs to be calculated. The updated state is also known as the estimated state.

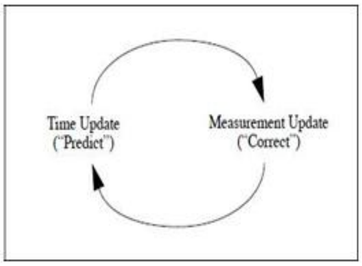

Fig. 4. Kalman Filter Cycle

A. Flowchart

The flowchart includes the various stages of the Kalman filter channel estimation used in the project done. The first stage is the assumption of a communication channel for the transmission. Here a Rayleigh fading channel is considered. Then the initial parameters for the estimation are assumed so that the prediction and updation in estimation can be started.

The estimation process is started and the updated and predicted estimates are found out. The errors and the total harmonic distortion are calculated then. Since the THD for normal house hold purposes is less than 3%. But in the case of fading the accepted fading is 15%. So when THD becomes within the 15% value, then the iteration is automatically stopped and the graphs are plotted.

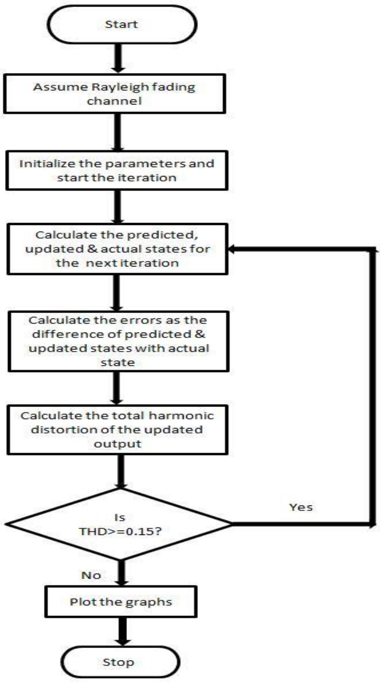

Fig 5. Flowchart

B. Algorithm

1. Start
2. Channel with Rayleigh fading is assumed.
3. Following parameters are assumed: number of iterations (N), state error (W), measurement error (V), measurement state (H), process noise covariance (Q), measurement noise covariance (R), updated estimates, predicted estimates, actual state (x), Kalman gain (K).
4. Updated state, predicted state and error covariance estimates, the actual state and Kalman gain for N iterations using the EKF equations as in [7].
5. Error1 and error2 as the difference of updated and predicted state with the actual state Using the equations given below are calculated.
   error1 = actual state – predicted state
   error2 = actual state –updated state
6. Total harmonic distortion(THD) of the updated sine output as the ratio of sum of harmonics power to the fundamental power in both of the channels is found.
7. If THD >= 0.15, go to step 4, else go to step 8.
8. Simulation results are plotted.
9. end

IV. SIMULATION RESULTS

Actual state, updated state and predicted state have been calculated. Then the error and THD also are

calculated, the iteration is stopped when the THD come inside 20% which is expected value.

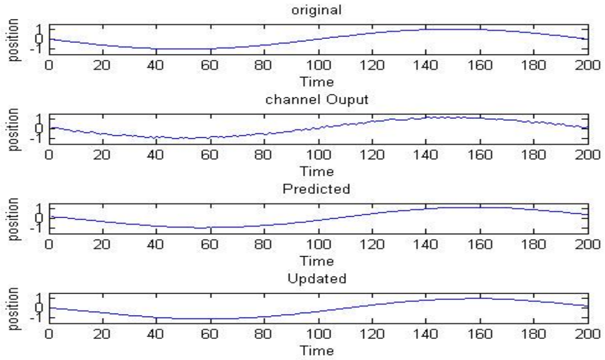

Fig. 6. Original state, channel o/p state, predicted state and updated state.

The original state is the actual sine wave which is to be received as per the equation of the Kalman based Filter algorithm. The predicted state and the estimated/updated state are also shown in the Fig. 6.

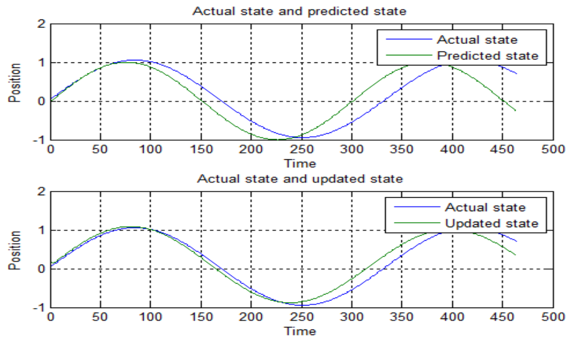

Fig. 7. The combined figure of the actual, predicted and updated state.

Fig. 7 makes it possible to exactly find the difference present in between the actual state, predicted state and the updated state. The actual wave is the actual state that had to be received at the receiver. The distortions indicate the error or fault associated with the Kalman based algorithm estimation method.

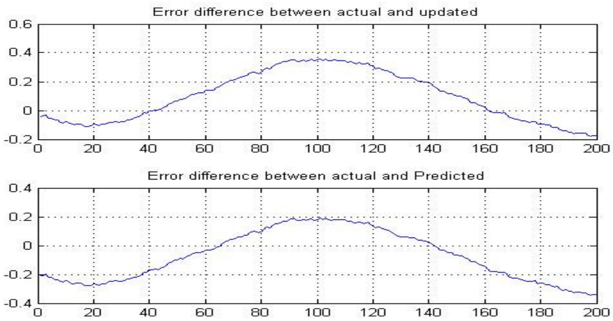

Figure 8. The error between the predicted and updated state with the actual state.

Fig. 8 shows the error between the actual state and the predicted state and the error between the actual state and the updated state. It is clear that the error by the Kalman based algorithm estimation method is comparatively less.

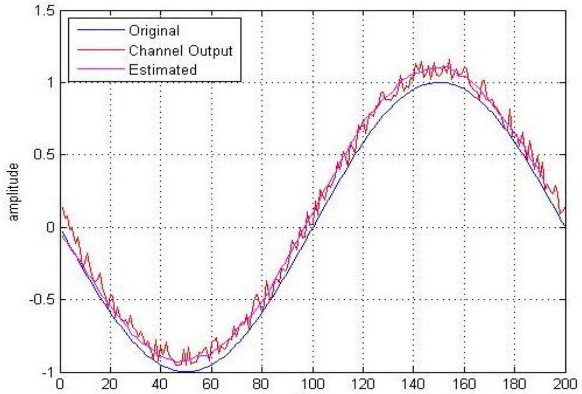

Fig. 9. Original signal Channel output, and estimated o/p.

Fig. 9 shows the original sine wave, channel output and the estimated channel output of the system.

V. CONCLUSIONS

In this paper, Kalman filter based algorithm has been implemented and the distortion of the updated estimated sine wave is calculated in a Rayleigh fading environment. Through simulation results it was found that the distortion value is limited below 15% by increasing number of iterations. Further, the total harmonic distortion converged is below 15% at the 464th iteration of the estimation process as per the assumed initial values considered in this process.

REFERENCES

[1] Gabriel Terejanu, Extended Kalman, Department of Computer Science & Engineering, University at Buffalo, NY 14260.

[2] Rupul Safaya, "A Multipath Channel Estimation Algorithm using a Kalman filter"

[3] Steven M. Kay, Fundamentals of Statistical Signal Processing, Volume I: Estimation Theory (v. 1).

[4] Vincent Poor, An Introduction to Signal Detection and Estimation, John Wiley, NY.

[5] Greg Welch & Gary Bishop, "An Introduction to Kalman Filter" , TR 95-041,Department of Computer Science, University of North Carolina, Chapel Hill, Chapel Hill-NC-27599-3175, 24 July, 2006.

[6] Rachel Kleinbauer, "Kalman filtering Implementation with MATLAB", University at Stuttgart.

[7] Maria Isabel Ribeiro, "Kalman and Extended Kalman Filters: Concept, Derivation and Properties" , Institute for Systems and Robotics.

[8] T.S. Rappaport, Wireless Communications Principles and Practice.. Publisher: Prentice Hall press. 1996.